SOUND FIELD CORRECTING METHOD IN AUDIO SYSTEM

BACKGROUND OF THE INVENTION

The present invention relates to a sound field correcting method of correcting a sound field characteristic in an audio system.

The audio system is required to produce the proper sound field space that can give a presence. In the prior art, the sound field correcting method in the audio system disclosed in the Japanese Utility Model Application Publication (KOKAI) Hei 6-13292 has been known.

The audio system in the prior art is the so-called multi-channel audio system having loudspeakers for a plurality of channels, and an equalizer for adjusting frequency characteristics of input audio signals and delay circuits for delaying the audio signals output from the equalizer are provided, and then outputs of the delay circuits on respective channels are supplied to respective loudspeakers on a plurality of channels.

Also, in order to correct the sound field characteristic, there are provided a pink noise generator, an impulse generator, a selector circuit, a microphone used to measure the reproduced sound being reproduced by the loudspeakers, a frequency analyzing means, and a delay time calculating means. Then, a pink noise generated by the pink noise generator is supplied

to the equalizer via the selector circuit on respective channels, and an impulse signal generated by the impulse generator is directly supplied to the loudspeakers on respective channels via the selector circuit.

Upon correcting the phase characteristic of the sound field space, propagation delay times of the impulse sound from the loudspeakers to a listening position are measured by measuring the impulse sound reproduced via the loudspeakers by the microphone while supplying directly the impulse signal from the above impulse generator to the loudspeakers and then analyzing the measured signals by using the delay time calculating means.

In other words, the propagation delay times of respective impulse sounds are measured by directly supplying the impulse signal to respective loudspeakers and calculating time differences from points of time when respective impulse signals are supplied to respective loudspeakers to points of time when respective impulse sounds being reproduced by every loudspeaker come up to the microphone by using the delay time calculating means. Thus, the phase characteristic of the sound field space can be corrected by adjusting the delay times of respective channels of the above delay circuit based on respective measured propagation delay times.

Also, upon correcting the frequency characteristic of the sound field space, the pink noise is supplied from the pink

noise generator to the equalizer on respective channels and then respective reproduced sounds of the pink noise reproduced via respective loudspeakers are measured by the microphone, and then frequency characteristics of the measured signals are analyzed by the frequency analyzing means. Thus, the frequency characteristic of the sound field space can be corrected by feedback-controlling the frequency characteristics of the equalizers on respective channels based on the analyzed results.

However, in the sound field correcting method in the audio system in the prior art, since levels (sound pressures) of respective reproduced sounds reproduced by a plurality of loudspeakers are not adjusted between the channels, such a phenomenon occurs that levels of the reproduced sounds reproduced by a low frequency band exclusively reproducing loudspeaker and all frequency band reproducing loudspeakers are enhanced in the low frequency when the sound field correction of the multi-channel audio system, that has the low frequency band exclusively reproducing loudspeaker such as a subwoofer and the all frequency band reproducing loudspeakers which can reproduce the audio signals over the overall audio frequency band, for example, is carried out. Therefore, the problems are caused such that the faithful audio reproduction cannot be achieved and thus this gives the unpleasant feeling to the listener, etc.

SUMMARY OF THE INVENTION

It is an object of the present invention to overcome the above subjects in the prior art and provide an automatic sound field correcting system capable of providing a higher quality sound field space.

A sound field correcting method in an audio system of the present invention, for supplying audio signals to a first sound generating means having a first reproducing frequency band and a second reproducing frequency band and a second sound generating means having the second reproducing frequency band respectively to reproduce them, the correcting method comprising a first step of supplying a noise to the first sound generating means and then detecting a reproduced sound in the first reproducing frequency band and a reproduced sound in the second reproducing frequency band, that are reproduced by the first sound generating means; a second step of supplying the noise to the second sound generating means and then detecting the reproduced sound in the second reproducing frequency band; and a third step of adjusting levels of the audio signals supplied to the first sound generating means and the second sound generating means such that a sum of a spectrum average level of the reproduced sound in the second reproducing frequency band reproduced by the first sound generating means and detected by the first step and a spectrum average level

of the reproduced sound in the second reproducing frequency band reproduced by the second sound generating means and detected by the second step and a spectrum average level of the reproduced sound in the first reproducing frequency band detected by the first step are set equal to a ratio of predetermined target characteristics.

Also, a sound field correcting method in an audio system of the present invention, for supplying audio signals to a first sound generating means having a first reproducing frequency band and a second reproducing frequency band and a second sound generating means having the second reproducing frequency band respectively to reproduce them, the correcting method comprising a first step of supplying a noise to the first sound generating means and then detecting a reproduced sound in the first reproducing frequency band and a reproduced sound in the second reproducing frequency band, that are reproduced by the first sound generating means; a second step of supplying the noise to the second sound generating means and then detecting the reproduced sound in the second reproducing frequency band; and a third step of adjusting levels of the audio signals supplied to the first sound generating means and the second sound generating means such that a ratio of a sum of a spectrum average level of the reproduced sound in the second reproducing frequency band reproduced by the first sound generating means and detected by the first step and a spectrum average level

of the reproduced sound in the second reproducing frequency band reproduced by the second sound generating means and detected by the second step to a spectrum average level of the reproduced sound in the first reproducing frequency band detected by the first step is set equal to a predetermined value.

According to such sound field correcting method, it is possible to make the levels of the reproduced sounds reproduced by the first sound generating means and the second sound generating means flat over the overall audio frequency band. As a result, the problems to give the unpleasant feeling to the listener, i.e., the levels of the reproduced sounds in the frequency band in which the first reproducing frequency band of the first sound generating means and the second reproducing frequency band of the second sound generating means are overlapped with each other are enhanced or weakened, can be overcome. Thus, the high quality sound field space with the presence can be implemented.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG.1 is a block diagram showing a configuration of an audio system including an automatic sound field correcting system according to the present embodiment.

FIG.2 is a block diagram showing a configuration of the automatic sound field correcting system.

FIG.3 is a block diagram showing a pertinent configuration of the automatic sound field correcting system according to the present embodiment.

FIG.4 is a block diagram showing another pertinent configuration of the automatic sound field correcting system.

FIG.5 is a view showing a frequency characteristic of a band- pass filter.

FIG.6 is a view showing the problem in a low frequency band of a reproduced sound.

FIG.7 is a view showing an example of arrangement of loudspeakers.

FIG.8 is a flowchart showing an operation of the automatic sound field correcting system.

FIG.9 is a flowchart showing a frequency characteristic correcting process.

FIG.10 is a flowchart showing a channel-to-channel level correcting process.

FIG.11 is a flowchart showing a delay characteristic correcting process.

FIG.12 is a flowchart showing a flatness correcting process.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

An embodiment of an automatic sound field correcting system to which a sound field correcting method of the present

invention is applied will be explained with reference to the accompanying drawings hereinafter. FIG.1 is a block diagram showing a configuration of an audio system including the automatic sound field correcting system to which a sound field correcting method according to the present embodiment is applied. FIG.2 to FIG.4 are block diagrams showing the configuration of the automatic sound field correcting system.

In FIG.1, a signal processing circuit 2 to which digital audio signals S_{FL}, S_{FR}, S_{C}, S_{RL}, S_{RR}, S_{WF} are supplied from a sound source 1 such as a CD (Compact Disk) player, a DVD (Digital Video Disk or Digital Versatile Disk) player, etc. via a signal transmission line having a plurality of channels, and a noise generator 3 are provided to the present audio system.

Also, D/A converters 4_{FL}, 4_{FR}, 4_{C}, 4_{RL}, 4_{RR}, 4_{WF} for converting digital outputs D_{FL}, D_{FR}, D_{C}, D_{RL}, D_{WF} which are signal-processed by the signal processing circuit 2 into analog signals, and amplifiers 5_{FL}, 5_{FR}, 5_{C}, 5_{RL}, 5_{RR}, 5_{WF} for amplifying respective analog audio signals being output from these D/A converters are provided. Respective analog audio signals SP_{FL}, SP_{FR}, SP_{C}, SP_{RL}, SP_{RR}, SP_{WF} amplified by these amplifiers are supplied to loudspeakers 5_{FL}, 5_{FR}, 5_{C}, 5_{RL}, 5_{RR}, 5_{WF} on a plurality of channels arranged in a listening room 7, etc., as shown in FIG.7, to sound them.

In addition, a microphone 8 for collecting reproduced sounds at a listening position RV, an amplifier 9 for amplifying

a sound collecting signal SM output from the microphone 8, and an A/D converter 10 for converting an output of the amplifier 9 into digital sound collecting data DM to supply to the signal processing circuit 2 are provided.

Then, the present audio system provides a sound field space with a presence to the listener at the listening position RV by sounding all frequency band type loudspeakers 6_{FL}, 6_{FR}, 6_{C}, 6_{RL}, 6_{RR} each has a frequency characteristic that enables an almost full range of the audio frequency band to reproduce, and a low frequency band exclusively reproducing loudspeaker 6_{WF} that has a frequency characteristic to reproduce only the so-called heavy and low sound.

For example, as shown in FIG.7, in the case that the listener arranges the front loudspeakers (front left-side loudspeaker, front right-side loudspeaker) 6_{FL}, 6_{FR} on two right and left channels and the center loudspeaker 6_{C} in front of the listening position RV, arranged the rear loudspeakers (rear left-side loudspeaker, rear right-side loudspeaker) 6_{RL}, 6_{RR} on two right and left channels at the rear of the listening position RV, and arranges the low frequency band exclusively reproducing subwoofer 6_{WF} at any position according to his or her taste, the automatic sound field correcting system installed in the present audio system can implement the sound field space with the presence by sounding six loudspeakers 6_{FL}, 6_{FR}, 6_{C}, 6_{RL}, 6_{RR}, 6_{WF} by supplying the analog audio signals SP_{FL},

SP_{FR}, SP_C, SP_{RL}, SP_{RR}, SP_{WF}, whose frequency characteristic and phase characteristic are corrected, to these loudspeakers.

The signal processing circuit 2 is composed of a digital signal processor (DSP), or the like. The automatic sound field correcting system consists of the digital signal processor (DSP), etc., that cooperate with the noise generator 3, the amplifier 9, and the A/D converter 10 to execute the sound field correction.

More particularly, system circuits CQT_1, CQT_2, CQT_3, CQT_4, CQT_5, CQT_k which are provided to signal transmission lines on respective channels shown in FIG.2 to have the almost similar configuration, a frequency characteristic correcting portion 11, a channel-to-channel level correcting portion 12, a phase characteristic correcting portion 13, and a flatness correcting portion 14 shown in FIG.3 are provided to the signal processing circuit 2. Then, the automatic sound field correcting system is constructed such that the frequency characteristic correcting portion 11, the channel-to-channel level correcting portion 12, the phase characteristic correcting portion 13, and the flatness correcting portion 14 can control the system circuits CQT_1, CQT_2, CQT_3, CQT_4, CQT_5, CQT_k. In this case, in the following explanation, respective channels are denoted by numbers x ($1 \leqq x \leqq k$).

A configuration of the system circuit CQT_1 provided to the first channel (x=1) will be explained on behalf of the

system circuits. Such configuration includes a switch element SW_{12} that ON/OFF-controls an input of the digital audio signal S_{FL} from the sound source 1 and a switch element SW_{11} that ON/OFF-controls an input of a noise signal DN from the noise generator 3. Also, the switch element SW_{11} is connected to the noise generator 3 via a switch element SW_N.

The switch elements SW_{11}, SW_{12}, SW_N are controlled by a system controller MPU that consists of a microprocessor described later. At the time of reproducing the audio sound, the switch element SW_{12} is turned ON (conductive) and the switch elements SW_{11}, SW_N are turned OFF (nonconductive). At the time of correcting the sound field, the switch element SW_{12} is turned OFF and the switch elements SW_{11}, SW_N are turned ON.

Band-pass filters BPF_{11} to BPF_{1j} are connected in parallel to output contacts of the switch elements SW_{11}, SW_{12} as frequency discriminating means, and thus the frequency dividing means that divides the frequency of the input signal is constructed by the overall band-pass filters BPF_{11} to BPF_{1j}.

In this case, suffixes 11 to 1j attached to BPF_{11} to BPF_{1j} denote the order of center frequencies f1 to fj of the band-pass filters BPF_{11} to BPF_{1j} on the first channel (x=1).

Attenuators ATF_{11} to ATF_{1j} being called an inter-band attenuator are connected to output contacts between the band-pass filters BPF_{11} to BPF_{1j} respectively. Accordingly, the attenuators ATF_{11} to ATF_{1j} act as an in-channel level

adjusting means that adjusts respective output levels of the band-pass filters BPF_{11} to BPF_{1j}.

Also, respective inter-band attenuators ATF_{11} to ATF_{1j} are provided to respective band-pass filters BPF_{11} to BPF_{1j} correspondingly, and variable gain type frequency discriminating means are composed of the band-pass filters and the inter-band attenuators, that correspond to each other, respectively. That is, BPF_{11} and ATF_{11} constitute the first variable gain type frequency discriminating means, BPF_{12} and ATF_{12} constitute the second variable gain type frequency discriminating means, ..., BPF_{1j} and ATF_{1j} constitute the j-th variable gain type frequency discriminating means.

Also, an adder ADD_1 is connected to output contacts of the inter-band attenuators ATF_{11} to ATF_{1j}, an attenuator ATG_1 being called a channel-to-channel attenuator is connected to an output contact of the adder ADD_1, and a delay circuit DLY_1 is connected to an output contact of the channel-to-channel attenuator ATG_1. Then, an output D_{FL} of the delay circuit DLY_1 is supplied to the D/A converter 4_{FL} shown in FIG.1.

Then, as shown in the frequency characteristic diagram of FIG.5, the band-pass filters BPF_{11} to BPF_{1j} are formed by narrow band passing type secondary Butterworth filters whose center frequencies are set to f1, f2,...fi,...fj respectively.

In other words, the band-pass filters BPF_{11} to BPF_{1j} that have frequencies f1, f2,...fi,...fj as center frequencies

respectively are provided. Such frequencies f1, f2,...fi,...fj are previously decided by dividing all frequency band of the loudspeaker 6_{FL}, that can reproduce over the low frequency band to the middle/high frequency band, by any number j. More particularly, the low frequency band that is less than about 0.2 kHz is divided into about six ranges and also the middle/high frequency band that is more than about 0.2 kHz is divided into about seven ranges, and then the center frequencies of respective divided narrow frequency ranges are set as the center frequencies f1, f2,...fi,...fj of the band-pass filters BPF_{11} to BPF_{1j}. In addition, all frequency bands are covered without omission by setting the center frequencies not to form clearances between respective passing frequency bands of the band-pass filters BPF_{11} to BPF_{1j} and not to overlap substantially respective passing frequency bands.

Also, exclusive ON/OFF switching of the band-pass filters BPF_{11} to BPF_{1j} can be performed mutually under the control of the system controller MPU. Also, in reproducing the audio sound, all band-pass filters BPF_{11} to BPF_{1j} are switched into their conductive states.

The attenuators ATF_{11} to ATF_{1j} consist of a digital attenuator respectively, and changes their attenuation factors in the range of 0 dB to the (-) side in accordance with adjust signals SF_{11} to SF_{1j} supplied from the frequency characteristic correcting portion 11.

The adder ADD1 adds signals that are passed through the band-pass filters BPF_{11} to BPF_{1j} and attenuated by the attenuators ATF_{11} to ATF_{1j} and then supplies the added signal to the attenuator ATG_1.

The channel-to-channel attenuator ATG_1 consists of the digital attenuator. Although its details will be given in the explanation of operation, the channel-to-channel attenuator ATG_1 changes its attenuation factor in the range of 0 dB to the (-) side in compliance with the adjust signal SG_1 from the channel-to-channel level correcting portion 12.

The delay circuit DLY_1 consists of the digital delay circuit, and changes its delay time in compliance with the adjust signal SDL_1 supplied from the phase characteristic correcting portion 13.

Then, the system circuits CQT_2, CQT_3, CQT_4, CQT_5 on remaining channels x=2 to 5 have a similar configuration to the system circuit CQT_1.

More particularly, although shown simply in FIG.2, following to the switch elements SW_{21}, SW_{22}, j variable gain type frequency discriminating means consisting of j band- pass filters BPF_{21} to BPF_{2j} that are set to the above center frequencies f1 to fj and inter-band attenuators ATF_{21} to ATF_{2j} that change their attenuation factors in the range of 0 dB to the (-) side in compliance with adjust signals SF_{21} to SF_{2j} supplied from the frequency characteristic correcting portion

11 is provided to the system circuits CQT_2 on the second channel (x=2). In addition, an adder ADD_2, an channel-to-channel attenuator ATG_2 for changing its attenuation factor in the range of 0 dB to the (-) side in compliance with an adjust signal SG_2 supplied from the channel-to-channel level correcting portion 12, and a delay circuit DLY_2 for changing its delay time in compliance with an adjust signal SDL_2 supplied from the phase characteristic correcting portion 13 are provided.

Following to the switch elements SW_{31}, SW_{32}, j variable gain type frequency discriminating means consisting of j band-pass filters BPF_{31} to BPF_{3j} that are set to the above center frequencies f1 to fj and inter-band attenuators ATF_{31} to ATF_{3j} is provided to the system circuits CQT_3 on the third channel (x=3). In addition, an adder ADD_3, an channel-to- channel attenuator ATG_3, and a delay circuit DLY_3 are provided. Then, like the system circuit CQT_1, the inter-band attenuators ATF_{31} to ATF_{3j}, the channel- to-channel attenuator ATG_3, and the delay circuit DLY_3 are adjusted respectively in compliance with adjust signals SF_{31} to SF_{3j} supplied from the frequency characteristic correcting portion 11, an adjust signal SG_3 supplied from the channel-to-channel level correcting portion 12, and an adjust signal SDL_3 supplied from the phase characteristic correcting portion 13.

Following to the switch elements SW_{41}, SW_{42}, j variable gain type frequency discriminating means consisting of j

band-pass filters BPF_{41} to BPF_{4j} that are set to the above center frequencies f1 to fj and inter-band attenuators ATF_{41} to ATF_{4j} is provided to the system circuits CQT_4 on the fourth channel (x=4). In addition, an adder ADD_4, an channel-to- channel attenuator ATG_4, and a delay circuit DLY_4 are provided. Then, like the system circuit CQT_1, the inter- band attenuators ATF_{41} to ATF_{4j}, the channel-to-channel attenuator ATG_4, and the delay circuit DLY_4 are adjusted respectively in compliance with adjust signals SF_{41} to SF_{4j} supplied from the frequency characteristic correcting portion 11, an adjust signal SG_4 supplied from the channel- to-channel level correcting portion 12, and an adjust signal SDL_4 supplied from the phase characteristic correcting portion 13.

Following to the switch elements SW_{51}, SW_{52}, j variable gain type frequency discriminating means consisting of j band-pass filters BPF_{51} to BPF_{5j} that are set to the above center frequencies f1 to fj and inter-band attenuators ATF_{51} to ATF_{5j} is provided to the system circuits CQT_5 on the fifth channel (x=5). In addition, an adder ADD_5, an channel-to- channel attenuator ATG_5, and a delay circuit DLY_5 are provided. Then, like the system circuit CQT_1, the inter- band attenuators ATF_{51} to ATF_{5j}, the channel-to-channel attenuator ATG_5, and the delay circuit DLY_5 are adjusted respectively in compliance with adjust signals SF_{51} to SF_{5j} supplied from the frequency characteristic correcting portion 11, an adjust signal SG_5

supplied from the channel- to-channel level correcting portion 12, and an adjust signal SDL_5 supplied from the phase characteristic correcting portion 13.

However, the system circuit CQTk on the sixth subwoofer channel (x=k) is constructed such that i (i<j) band-pass filters BPF_{k1} to BPF_{kj}, that pass only divided low frequency bands (frequencies below about 0.2 kHz) shown in FIG.5 respectively, and inter-band attenuators ATF_{k1} to ATF_{kj} are connected in parallel following to the switch elements SW_{k1}, SW_{k2}, then an adder ADD_k adds outputs of the attenuators ATF_{k1} to ATF_{ki}, then an output of the added result is passed through a channel-to-channel attenuator ATG_k and a delay circuit DLY_k, and then an output D_{WF} of the delay circuit DLY_k is supplied to the D/A converter 4_{WF}.

In this case, i variable gain type frequency discriminating means consist of band-pass filters BPF_{k1} to BPF_{ki} and inter-band attenuators ATF_{k1} to ATF_{ki}.

Next, in FIG.3, the frequency characteristic correcting portion 11 receives respective sound collecting data DM obtained when the loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR}, 6_{WF} are sounded individually by the noise signal (pink noise) DN output from the noise generator 3, and then calculates levels of the reproduced sounds of respective loudspeakers at the listening position RV based on the sound collecting data DM. Then, the frequency characteristic correcting portion 11 generates the

adjust signals SF_{11} to SF_{1j}, SF_{21} to SF_{2j}, ..., SF_{k1} to SF_{ki} based on these calculated results to correct automatically the attenuation factors of the inter-band attenuators ATF_{11} to ATF_{1j}, ATF_{21} to ATF_{2j}, ..., ATF_{k1} to ATF_{ki} individually.

Based on the above correction of the attenuation factors by the frequency characteristic correcting portion 11, gain adjustment for respective passing frequencies of the band-pass filters BPF_{11} to BPF_{ki} provided to the system circuits CQT_1 to CQT_k is carried out every channel.

That is, the frequency characteristic correcting portion 11 adjusts the levels of respective signals output from the band-pass filters BPF_{11} to BPF_{ki} by performing the gain adjustment of the inter-band attenuators ATF_{11} to ATF_{ki} serving as an in-channel level adjusting means, whereby the frequency characteristic correcting portion 11 acts as an in-channel level correcting means for setting the frequency characteristic.

The channel-to-channel level correcting portion 12 receives respective sound collecting data DM obtained when all frequency band loudspeakers 6_{FL}, 6_{FR}, 6_{C}, 6_{RL}, 6_{RR} are sounded individually by the noise signal (pink noise) DN output from the noise generator 3, and then calculates the levels of the reproduced sounds of respective loudspeakers at the listening position RV based on the sound collecting data DM. Then, the channel-to-channel level correcting portion 12 generates the

adjust signals SG_1 to SG_5 based on these calculated results and corrects automatically the attenuation factors of the channel-to-channel attenuators ATG_1 to ATG_5 by the adjust signals SG_1 to SG_5.

Based on the correction of the attenuation factors by the channel-to-channel level correcting portion 12, the level adjustment (gain adjustment) between the system circuits CQT_1 to CQT_5 on the first to fifth channels is carried out.

That is, the channel-to-channel level correcting portion 12 acts as a channel-to-channel level correcting means that corrects levels of the audio signals transmitted every channel (signal transmission line) between channels.

However, the channel-to-channel level correcting portion 12 does not adjust the attenuation factor of the channel-to-channel attenuator ATG_k provided to the system circuit CQT_k on the subwoofer channel, but the flatness correcting portion 14 adjusts the attenuation factor of the channel-to-channel attenuator ATG_k.

The phase characteristic correcting portion 13 measures the phase characteristic of respective channels based on respective sound collecting data DM obtained when respective loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR}, 6_{WF} are sounded individually by supplying the noise signal (uncorrelated noise) DN output from the noise generator 3 to the system circuits CQT_1 to CQT_k on respective channels, and then corrects the phase

characteristic of the sound field space in compliance with the measured result.

More particularly, the loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR}, 6_{WF} on respective channels are sounded by the noise signal DN every period T, and then cross correlations between resultant sound collecting data DM_1, DM_2, DM_3, DM_4, DM_5, DM_k on respective channels are calculated. Here, the cross correlation between the sound collecting data DM_2 and DM_1, the cross correlation between the sound collecting data DM_3 and DM_1, ..., the cross correlation between the sound collecting data DM_k and DM_1 are calculated, and then peak intervals (phase differences) between respective correlation values are set as their delay times $\tau 2$ to τk in respective system circuits CQT_2 to CQT_k. That is, the delay times $\tau 2$ to τk of remaining system circuits CQT_2 to CQT_k are calculated on the basis of the phase of the sound collecting data DM1 obtained from the system circuit CQT_1 (i.e., phase difference 0, $\tau 1=0$). Then, the adjust signals SDL_1 to SDL_k are generated based on measured results of these delay times $\tau 2$ to τk, and then the phase characteristic of the sound field space is corrected by automatically adjusting respective delay times of the delay circuits DLY_1 to DLY_k by using these adjust signals SDL_1 to SDL_k. In this case, the uncorrelated noise is employed to correct the phase characteristic in the present embodiment, but either the pink noise or other noise may be employed.

The flatness correcting portion 14 adjusts the attenuation factor of the channel-to-channel attenuator ATG_k in the system circuit CQT_k, that is not adjusted by the channel-to-channel level correcting portion 12, after the adjustments made by the frequency characteristic correcting portion 11, the channel-to-channel level correcting portion 12, and the phase characteristic correcting portion 13 have been completed.

That is, as shown in FIG.4, the flatness correcting portion 14 comprises a middle/high frequency band processing portion 15a, a low frequency band processing portion 15b, a subwoofer low frequency band processing portion 15c, and a calculating portion 15d.

In the state that the low frequency band-pass filters BPF_{11} to BPF_{1i}, BPF_{21} to BPF_{2i}, BPF_{31} to BPF_{3i}, BPF_{41} to BPF_{4i}, BPF_{51} to BPF_{5i} provided to the system circuits CQT1 to CQT5 are turned OFF and the remaining middle/high frequency band-pass filters are turned ON, the middle/high frequency band processing portion 15a measures a spectrum average level P_{MH} of the reproduced sound in the middle/high frequency band from the sound collecting data DM (referred to as "middle/high frequency band sound collecting data D_{MH}" hereinafter) that are obtained when all frequency band loudspeakers 6_{FL}, 6_{FR}, 6_{C}, 6_{RL}, 6_{RR} are sounded simultaneously based on the noise signal (uncorrelated noise) DN output from the noise generator 3.

In the state that the low frequency band-pass filters BPF_{11} to BPF_{1i}, BPF_{21} to BPF_{2i}, BPF_{31} to BPF_{3i}, BPF_{41} to BPF_{4i}, BPF_{51} to BPF_{5i} provided to the system circuits CQT_1 to CQT_5 are turned ON and the remaining middle/high frequency band- pass filters are turned OFF, the low frequency band processing portion 15b measures a spectrum average level P_L of the reproduced sound in the low frequency band from the sound collecting data DM (referred to as "low frequency band sound collecting data D_L" hereinafter) that are obtained when all frequency band loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR} are sounded simultaneously based on the noise signal (uncorrelated noise) DN output from the noise generator 3.

In the condition that all band-pass filters BPF_{k1} to BPF_{ki} provided to the system circuit CQT_k on the subwoofer channel are turned ON, the low frequency band processing portion 15c measures a spectrum average level P_{WFL} of the low sound reproduced only by the loudspeaker 6_{WF} from the sound collecting data DM (referred to as "subwoofer sound collecting data D_{WFL}" hereinafter) that are obtained when the low frequency exclusively reproducing loudspeaker 6_{WF} is sounded based on the noise signal (pink noise) DN output from the noise generator 3.

The calculating portion 15d generates the adjust signal SG_k that makes the frequency characteristic of the reproduced sound at the listening position RV flat over all audio frequency

bands when all loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR}, 6_{WF} are sounded simultaneously, by executing predetermined calculating processes explained later in detail based on the spectrum average level P_{MH} in the above middle/high frequency band and the spectrum average levels P_L, P_{WFL} in the low frequency bands.

That is, as shown in the frequency characteristic diagram of FIG.6, since the all frequency band loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR} have not only the middle/high frequency band reproducing capability but also the low frequency band reproducing capability, in some cases the total spectrum average level of the low frequency sounds reproduced by the loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR} and the low frequency sound reproduced by the loudspeaker 6_{WF}, for example, become higher than the spectrum average level of the reproduced sound in the middle/high frequency band if these loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR} and the low frequency band exclusively reproducing loudspeaker 6_{WF} are sounded. Thus, there is caused such a problem that such low frequency sounds are offensive to the ear and also give the listener an unpleasant feeling. Therefore, the calculating portion 15d adjusts the attenuation factor of the channel-to-channel attenuator ATG_k by the adjust signal SG_k such that the total spectrum average level of the above low frequency sounds and the spectrum average level of the middle/high frequency sounds can be made flat.

Accordingly, the flatness correcting portion 14 as well

as the channel-to-channel level correcting portion 12 acts as the channel-to-channel level correcting means that corrects the levels of the audio signals transmitted every channel (signal transmission line) between the channels.

In this case, the configuration of the automatic sound field correcting system is explained, but more detailed functions will be explained in detail in the explanation of operation.

Next, an operation of the automatic sound field correcting system having such configuration will be explained with reference to flowcharts shown in FIG.8 to FIG.12 hereunder.

When the listener arranges a plurality of loudspeakers 6_{FL} to 6_{WF} in the listening room 7, etc. and connects them to the present audio system, as shown in FIG.7, for example, and then instructs to start the sound field correction by operating a remote controller (not shown) provided to the present audio system, the system controller MPU operates the automatic sound field correcting system in compliance with this instruction.

First, an outline of the operation of the automatic sound field correcting system will be explained with reference to FIG.8. In the frequency characteristic correcting process in step S10, the process for adjusting the attenuation factors of all inter-band attenuators ATF_{11} to ATF_{kj} provided to the system circuits CQT_1, CQT_2, CQT_3, CQT_4, CQT_5, CQT_k is carried

out by the frequency characteristic correcting portion 11.

Then, in the channel-to-channel level correcting process in step S20, the process for adjusting the attenuation factors of the channel-to-channel attenuators ATG_1 to ATG_5 provided to the system circuits CQT_1, CQT_2, CQT_3, CQT_4, CQT_5 is carried out by the channel-to-channel level correcting portion 12. That is, in step S20, the channel-to-channel attenuator ATG_k provided to the system circuit CQT_k on the subwoofer channel is not adjusted.

Then, in the phase characteristic correcting process in step S30, the process for adjusting the delay times of all delay circuits DLY_1 to DLY_k provided to the system circuits CQT_1, CQT_2, CQT_3, CQT_4, CQT_5, CQT_k is carried out by the phase characteristic correcting portion 13. That is, the process for correcting the phase characteristic of the reproduced sound being reproduced by all loudspeakers 6_{FL} to 6_{WF} is performed.

Then, in the flatness correcting process in step S40, the process for making the frequency characteristic of the reproduced sound at the listening position RV flat over the full audio frequency band is carried out by the flatness correcting portion 14.

In this manner, the present automatic sound field correcting system executes the sound field correction by performing in sequence the correcting processes that are roughly classified into four stages.

Then, respective processes in steps S10 to S40 will be explained in sequence.

First, the frequency characteristic correcting process in step S10 will be explained in detail. The process in step S10 will be carried out in compliance with the detailed flowchart shown in FIG.9.

In step S100, the initialization process is executed to set the attenuation factors of all inter-band attenuators ATF_{11} to ATF_{ki} and the channel-to-channel attenuators ATG_1 to ATG_k in the system circuits CQT_1, CQT_2, CQT_3, CQT_4, CQT_5, CQT_k shown in FIG.2 to 0 dB. Also, the delay times in all delay circuits DLY_1 to DLY_k are set to 0, and the amplification factors of the amplifiers 5_{FL} to 5_{WF} shown in FIG.1 are set equal.

In addition, the switch elements SW_{12}, SW_{22}, SW_{32}, SW_{42}, SW_{52}, SW_{k2} are turned OFF (nonconductive) to cut off the input from the sound source 1, and the switch elements SW_N is turned ON (conductive). Accordingly, the signal processing circuit 2 is set to the state that the noise signal (pink noise) DN generated by the noise generator 3 is supplied to the system circuits CQT_1, CQT_2, CQT_3, CQT_4, CQT_5, CQT_k.

Then, the process goes to step S102, flag data n=0 is set in a flag register (not shown) built in the system controller MPU.

Then, the sound field characteristic measuring process is executed in step S104.

In this step S104, the noise signal DN is supplied in sequence to the system circuits CQT_1 to CQT_k by exclusively turning ON the switch elements SW_{11}, SW_{21}, SW_{31}, SW_{41}, SW_{51}, SW_{k1} for the predetermined period T respectively. Also, the band-pass filters in the system circuit to which the noise signal DN is being supplied are exclusively turned ON in sequence from the low frequency band side to the middle/high frequency band side.

Accordingly, the noise signal DN that is frequency-divided by the band-pass filters BPF_{11} to BPF_{1j} in the system circuit CQT_1 is supplied to the loudspeaker 6_{FL} sequentially. As a result, the microphone 8 collects the noise sound that is produced at the listening position RV and is frequency-divided, and the D/A converter 10 supplies these sound collecting data DM (referred to as "DM_{11} to DM_{1j}" hereinafter) to the frequency characteristic correcting portion 11. Then, the frequency characteristic correcting portion 11 stores these sound collecting data DM_{11} to DM_{1j} in a predetermined memory portion (not shown).

Also, similarly the noise signal DN that is subjected to the frequency division is supplied to the loudspeakers 6_{FR} to 6_{WF} via remaining system circuits CQT_2 to CQT_k, and then resultant sound collecting data DM (referred to as "DM_{21} to DM_{2j}, DM_{31} to DM_{3j}, DM_{41} to DM_{4j}, DM_{51} to DM_{5j}, DM_{k1} to DM_{ki}" hereinafter) on respective channels are stored in the predetermined memory

portion (not shown).

In this manner, the sound collecting data [DAxJ] expressed by a matrix in Eq.(1) are stored in the frequency characteristic correcting portion 11 by executing the sound field characteristic measuring process. In this case, a suffix x in [DAxJ] denotes the channel number ($1 \leqq x \leqq k$), and a suffix J denotes the order of the center frequencies f1 to fj from the low frequency band to the middle/high frequency band.

$$[DAxJ] = \begin{bmatrix} DM11 & \cdots\cdots\cdots & DM1j \\ DM21 & \cdots\cdots\cdots & DM2j \\ DM31 & \cdots\cdots\cdots & DM3j \\ DM41 & \cdots\cdots\cdots & DM4j \\ DM51 & \cdots\cdots\cdots & DM5j \\ DMk1 & \cdots\cdots\ DMki & \end{bmatrix} \quad \cdots (1)$$

In addition, in step S104, the sound collecting data [DAxJ] are compared with predetermined threshold value THD_{CH} every channel, and sizes of the loudspeakers 6_{FL} to 6_{WF} on respective channels are decided based on the comparison results. That is, since the sound pressure of the reproduced sound reproduced by the loudspeaker is changed according to the size of the loudspeaker, the sizes of the loudspeakers on respective channels are decided.

As the concrete deciding means, if the size of the loudspeaker 6_{FL} on the first channel (x=1) is decided, an average

value of the sound collecting data DM_{11} to DM_{1j} on the first channel in above Eq.(1) is compared with the threshold value THD_{CH}. If the average value is smaller than the threshold value THD_{CH}, the loudspeaker 6_{FL} is decided as the small loudspeaker. Then, if the average value is larger than the threshold value THD_{CH}, the loudspeaker 6_{FL} is decided as the large loudspeaker. In addition, the loudspeakers 6_{FR}, 6_{C}, 6_{RL}, 6_{RR}, 6_{WF} on remaining channels are similarly decided.

Then, in the channels in which the loudspeakers being decided as the small loudspeaker are connected, processes in steps S106 to S124 described in the following are not executed. The processes in steps S106 to S124 are applied only to the channels in which the loudspeakers being decided as the large loudspeaker are connected.

In order to facilitate the understanding of explanation, the processes in steps S106 to S124 will be explained under the assumption that all the loudspeakers 6_{FL}, 6_{FR}, 6_{C}, 6_{RL}, 6_{RR}, 6_{WF} are the large loudspeaker.

Then, in step S106, the listener sets target curve data [TGxJ] that are set previously in the present audio system into the frequency characteristic correcting portion 11. Where the target curve denotes the frequency characteristic of the reproduced sound that can suit the listener's taste. In the present audio system, in addition to the target curve used to generate the reproduced sound having the frequency

characteristic that is suitable for the classic music, various target curve data [TGxJ] used to generate the reproduced sounds having the frequency characteristics that are suitable for rock music, pops, vocal, etc. are stored in the system controller MPU. Also, these target curve data [TGxJ] consist of an aggregation of the data of the same number as the inter-band attenuators ATF_{11} to ATF_{ki}, as shown by a matrix in Eq. (2), and they can be selected every channel independently.

$$[TGxJ] = \begin{bmatrix} TG11 & \cdots\cdots\cdots & TG1j \\ TG21 & \cdots\cdots\cdots & TG2j \\ TG31 & \cdots\cdots\cdots & TG3j \\ TG41 & \cdots\cdots\cdots & TG4j \\ TG51 & \cdots\cdots\cdots & TG5j \\ TGk1 & \cdots\cdots \quad TGki & \end{bmatrix} \quad \cdots (2)$$

Then, the listener can select these target curves freely by operating predetermined operation buttons of a remote controller. Then, the system controller MPU sets the selected target curve data [TGxJ] onto the frequency characteristic correcting portion 11.

However, if the listener instructs the sound field correction without selection of the target curve, all data TG_{11} to TG_{ki} are set to a previously decided value, e.g., 1.

Then, in step S108, the frequency characteristic correcting portion 11 sets the number of the first channel (x=1) and the order of the first center frequency (J=1), and then

calculates the adjust values F0(1,1) to F0(1,j) by repeating processes in steps S110 to S114 to adjust the inter-band attenuators ATF_{11} to ATF_{1j}.

More particularly, if the first line data DM_{11} to DM_{1j} in the sound collecting data [DAxJ] given by above Eq.(1) and the first line data TG_{11} to TG_{1j} in the target curve data [TGAxJ] given by above Eq.(2) are applied to following Eq.(3) while changing the variable J between 1 to j in steps S112 and S114 after the flag data n is set to 0 and a variable x representing the channel is set to 1, the adjust values F0(1,1) to F0(1,j) of the inter-band attenuators ATF_{11} to ATF_{1j} corresponding to the first channel are calculated. However, if a value TGxJ/DMxJ calculated by Eq.(3) has a calculation error that is smaller than the predetermined threshold value THD, the value TGxJ/DMxJ is forcedly set to 0 to achieve the improvement in the adjust precision.

$$Fn(x,J)=TGxJ/DMxJ \qquad \ldots(3)$$

Then, in step S112, if it is decided that all adjusted values F0(1,1) to F0(1,j) of the inter-band attenuators ATF_{11} to ATF_{1j} on the first channel have been calculated, the process goes to step S116. Then, it is decided whether or not the adjusted values of all inter-band attenuators on the second to sixth channels (x=2 to k) have been calculated. If NO, the variable x is incremented by 1 and the variable j is set to 1 in step S118, and then the processes from step S110 to step

S116 are repeated. Then, if the calculation of the adjusted values of all inter-band attenuators is finished, the process goes to step S120.

Accordingly, the adjusted values [F0xJ] of all inter-band attenuators ATF11 to ATF1j represented by the matrix given by following Eq.(4) are calculated.

$$[F0xJ] = \begin{bmatrix} F0(1,1) \cdots\cdots F0(1,j) \\ F0(2,1) \cdots\cdots F0(2,j) \\ F0(3,1) \cdots\cdots F0(3,j) \\ F0(4,1) \cdots\cdots F0(4,j) \\ F0(5,1) \cdots\cdots F0(5,j) \\ F0(k,1) \cdots F0(k,i) \end{bmatrix} \quad \cdots (4)$$

Then, in step S120, the adjusted values [F0xJ] are normalized by executing the calculation represented by the matrix in following Eq.(5), and then resultant normalized adjusted values [FN0xJ] are set as new target curve data [TGxJ]=[FN0xJ]. That is, the target curve data [TGxJ] in above Eq.(2) are replaced with the normalized adjusted values [FN0xJ].

$$[FN0xJ] = \begin{bmatrix} F0(1,1)/F01max \cdots\cdots F0(1,j)/F01max \\ F0(2,1)/F02max \cdots\cdots F0(2,j)/F02max \\ F0(3,1)/F03max \cdots\cdots F0(3,j)/F03max \\ F0(4,1)/F04max \cdots\cdots F0(4,j)/F04max \\ F0(5,1)/F05max \cdots\cdots F0(5,j)/F05max \\ F0(k,1)/F0kmax \cdots F0(k,i)/F0kmax \end{bmatrix} \quad \cdots (5)$$

In this case, values F01max to F0kmax having a suffix "max" in Eq. (5) are maximum values of the adjusted values on respective channels x=1 to k when the flag data n is n=1.

Then, in step S122, it is decided whether or not the flag data n is 1. If NO, the flag data n is set to 1 in step S124, and then the processes from step S104 to S120 are repeated.

In this manner, the processes in step S104 and subsequent steps are repeated. In step S122, if it is decided that the flag data n is 1, the process goes to step S126. While, if the processes in step S104 and subsequent steps are repeated, the flag data n is set to n=1 and thus the calculations in above Eqs. (1) to (5) are executed once again. Thus, the normalized adjusted values [FN1xJ] in following Eq. (6) corresponding to above Eq. (5) are calculated.

$$[FN1xJ] = \begin{bmatrix} F1(1,1)/F11max \cdots\cdots F1(1,j)/F11max \\ F1(2,1)/F12max \cdots\cdots F1(2,j)/F12max \\ F1(3,1)/F13max \cdots\cdots F1(3,j)/F13max \\ F1(4,1)/F14max \cdots\cdots F1(4,j)/F14max \\ F1(5,1)/F15max \cdots\cdots F1(5,j)/F15max \\ F1(k,1)/F1kmax \cdots F1(k,i)/F1kmax \end{bmatrix} \quad \cdots (6)$$

Then, in step S126, adjust data [SFxJ] used to adjust the attenuation factors of all inter-band attenuators ATF_{11} to ATF_{1j}, . . . , ATF_{k1} to ATF_{ki} of the system circuits CQT_1 to CQT_k shown in Eq. (7) are calculated by multiplying the normalized adjusted values [FN0xJ] by the normalized adjusted values [FN1xJ] in

respective matrices.

$$[SFxJ] = \begin{bmatrix} SF11 & \cdots\cdots\cdots\cdots & SF1j \\ SF21 & \cdots\cdots\cdots\cdots & SF2j \\ SF31 & \cdots\cdots\cdots\cdots & SF3j \\ SF41 & \cdots\cdots\cdots\cdots & SF4j \\ SF51 & \cdots\cdots\cdots\cdots & SF5j \\ SFk1 & \cdots SFki & \end{bmatrix} \quad \cdots(7)$$

That is, a value SF11 on the first row and the first column of the matrix in Eq.(7) is calculated by multiplying a value F0(1,1)/F01max on the first row and the first column of the normalized adjusted values [FN0xJ] and [FN1xJ] shown in Eqs.(5)(6) by a F1(1,1)/F11max, and then a value SF21 on the second row and the first column of the matrix in Eq.(7) is calculated by multiplying a value F0(2,1)/F02max on the second row and the first column by a F1(2,1)/F12max. In the subsequent, adjust data [SFxj] used for the attenuation factor adjustment represented by the matrix in Eq.(7) are calculated by executing the similar calculation in the following.

Then, the attenuation factors if the inter-band attenuators ATF_{11} to ATF_{1j}, ..., ATF_{k1} to ATF_{ki} are adjusted according to respective adjust signals SF_{11} to SF_{1j}, ..., SF_{k1} to SF_{ki} based on the adjust data [SFxJ], and then the process goes to step S20 in FIG.8.

Also, in the foregoing sound field characteristic measuring process in step S104, if the channel in which the

small loudspeaker is connected is decided, the attenuation factors of the inter-band attenuators provided in the channels are adjusted to 0 dB, while the attenuation factors of the inter-band attenuators in the channels in which the large loudspeakers are connected are adjusted based on the adjust data [SFxJ].

In step S104, if it is decided that the loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR}, 6_{WF} on all channels are all small loudspeakers, the process goes directly to the processes from step S104 to step S126 without executing steps S106 to S124. In step S126, the attenuation factors of the inter- band attenuators on all channels are adjusted to 0 dB.

In this way, the frequency characteristics of respective channels are corrected by adjusting the attenuation factors of the inter-band attenuators ATF_{11} to ATF_{ki} by virtue of the frequency characteristic correcting portion 11. Thus, the frequency characteristic of the sound field space is made proper.

Also, in the sound field characteristic measuring process in step S104, since respective loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR}, 6_{WF} are sounded by the pink noise on time-division basis, the frequency characteristics and the reproducing capabilities of respective loudspeakers can be detected under the substantially same conditions when the sound field is produced based on the actual audio signals. Therefore, the

total correction of the frequency characteristic can be achieved while taking account of the frequency characteristics and the reproducing capabilities of respective loudspeakers.

Next, the channel-to-channel level correcting process in step S20 will be carried out in compliance with a flowchart shown in FIG.10.

First, the initialization process in step S200 is executed, and the noise signal DN from the noise generator 3 can be input by switching the switch elements SW_{11} to SW_{51}. At this time, the switch elements SW_{k1}, SW_{k2} on the subwoofer channel are turned OFF. Also, the attenuation factors of the channel-to-channel attenuators ATG_1 to ATG_k are set to 0 dB. In addition, the delay times of all delay circuits DLY_1 to DLY_5 are set to 0. Further, the amplification factors of the amplifiers 5_{FL} to 5_{WF} shown in FIG.1 are made equal.

Besides, the attenuation factors of the inter-band attenuators ATF_{11} to ATF_{1j}, ATF_{21} to ATF_{2j}, ..., ATF_{k1} to ATF_{ki}, are fixed to the state that they have been adjusted by the above frequency characteristic correcting process.

Then, in step S202, the variable x representing the channel number is set to 1. Then, in step S204, the sound field characteristic measuring process is executed. The processes in steps S204 to S208 are repeated until the sound field characteristic measurement of the channels 1 to 5 is completed.

Here, the noise signal (pink noise) is supplied in

sequence to the system circuits CQT_1 to CQT_5 by exclusively turning ON the switch elements SW_{11}, SW_{21}, SW_{31}, SW_{41}, SW_{51} for the predetermined period T respectively while fixing the band-pass filters BPF_{11} to BPF_{1j}, . . . , BPF_{51} to BPF_{5j} in the normal ON (conductive) state(steps S206, S208).

The microphone 8 collects respective reproduced sounds being reproduced by the loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR} by this repeating process. Then, resultant sound collecting data DM (=DM_1 to DM_5) on the first to fifth channels are stored in the memory portion (not shown) in the channel-to-channel level correcting portion 12. That is, the sound collecting data [DBx] represented by the matrix in following Eq.(8) are stored.

$$[DBx] = \begin{bmatrix} DM1 \\ DM2 \\ DM3 \\ DM4 \\ DM5 \end{bmatrix} \quad \cdots (8)$$

Then, after the measurement of the sound field characteristics on the first to fifth channels has been finished, the process goes to step S210. Then, one sound collecting data having the minimum value is extracted from the sound collecting data DM_1 to DM_5. Then, the extracted data is set to the target data TG_{CH} for the channel-to-channel level correction.

Then, in step S212, the attenuation factor adjusted values [SGx] of the channel-to-channel attenuators ATG_1 to ATG_5 given by following Eq.(9) are calculated by normalizing the matrix in above Eq.(8) based on the target data TG_{CH} for the channel-to-channel level correction. Then, in step S214, the attenuation factors of the channel-to-channel attenuators ATG_1 to ATG_5 are adjusted by using the adjust signals SG_1 to SG_5 based on the attenuation factor adjust signals [SGx].

$$[SGx] = \begin{bmatrix} SG1 \\ SG2 \\ SG3 \\ SG4 \\ SG5 \end{bmatrix} = \begin{bmatrix} DM1/TGCH \\ DM2/TGCH \\ DM3/TGCH \\ DM4/TGCH \\ DM5/TGCH \end{bmatrix} \quad \cdots (9)$$

With the above processes, except the subwoofer channel, the level adjustment between the first to fifth channels in which all frequency band loudspeakers are connected is completed. Subsequently, the process goes to step S30 in FIG.8.

In this fashion, the level characteristics of respective channels are made proper by correcting the attenuation factors of the channel-to-channel attenuators ATG_1 to ATG_k by virtue of the channel-to-channel level correcting portion 12. Thus, the levels of the reproduced sounds of respective loudspeakers at the listening position RV are set properly.

Also, in the sound field characteristic measuring process in step S204, since resultant reproduced sounds are collected by sounding the loudspeakers 6_{FL}, 6_{FR}, 6_{C}, 6_{RL}, 6_{RR} on time-division basis, the reproducing capabilities (output powers) of respective loudspeakers can be detected. Therefore, it is possible to achieve the total rationalization while taking account of the reproducing capabilities of respective loudspeakers.

Next, the phase characteristic correcting process in step S30 will be carried out in compliance with a flowchart shown in FIG.11.

First, the initialization process in step S300 is executed. The noise signal (uncorrelated noise) DN output from the noise generator 3 can be input by switching the switch elements SW_{11} to SW_{k2}. Also, the inter-band attenuator ATF_{11} to ATF_{ki} and the channel-to-channel attenuators ATG_{1} to ATG_{k} are fixed to have the already- adjusted attenuation factors as they are, and also the delay times of the delay circuits DLY_{1} to DLY_{k} are set to 0. Further, the amplification factors of the amplifiers 5_{FL} to 5_{WF} shown in FIG.1 are made equal.

Then, in step S302, the variable x representing the channel number is set to 1 and a variable AVG is set to 0. Then, in step S304, the sound field characteristic measuring process is carried out to measure the delay times. Then, the processes in steps S304 to S308 are repeated until the sound field

characteristic measurement of the first to k-th channels have been completed.

Here, the noise signal DN is supplied to the system circuits CQT_1 to CQT_k for every period T by exclusively turning ON the switch elements SW_{11}, SW_{21}, SW_{31}, SW_{41}, SW_{k1} for the predetermined period T respectively.

According to this repeating process, the continuous noise signal DN is supplied to the loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR}, 6_{WF} for the period T respectively, and then the microphone 8 collects respective reproduced sounds of the noise signal DN being reproduced for the period T respectively. In addition, the phase characteristic correcting portion 13 receives respective sound collecting data DM (referred to as "DM_1, DM_2, DM_3, DM_4, DM_5, DM_k" hereinafter) that are output from the A/D converter 10 for the period T respectively. In this event, since the high-speed sampling is performed for respective periods T by the A/D converter 10, these sound collecting data DM_1, DM_2, DM_3, DM_4, DM_5, DM_k constitute a plurality of sampling data respectively.

When this measurement has been completed, the process goes to step S310 wherein the phase characteristics of respective channels are calculated. Here, the cross correlation between the sound collecting data DM_2 and DM_1 is calculated and then a peak interval (phase difference) between resultant correlation values is set as a delay time $\tau 2$ in the

system circuit CQT_2. Also, the cross correlations between remaining sound collecting data DM_3 to DM_k and the sound collecting data DM_1 are calculated respectively, and then peak intervals (phase differences) between resultant correlation values is set as delay times $\tau 3$ to τk in the system circuits CQT_3 to CQT_k. That is, the delay times $\tau 2$ to τk in remaining system circuits CQT_2 to CQT_k are calculated on the basis of the phase of the sound collecting data DM_1 obtained from the system circuit CQT_1 (i.e., phase difference 0).

Then, the process goes to step S312 wherein the variable AVG is incremented by 1. Then, in step S314, it is decided whether or not the variable AVG reaches a predetermined value AVERAGE. If NO, the processes starting from step S304 are repeated.

Here, the predetermined value AVERAGE is a constant indicating the number of times of the repeating processes in steps S304 to S312. In the present embodiment, the predetermined value AVERAGE is set to AVERAGE=4.

The delay times $\tau 1$ to τk of the system circuit CQT_1 to CQT_k are calculated for every four circuits by repeating the four times measuring process in this manner. Then, in step S316, average values $\tau 1'$ to $\tau k'$ of every four delay times $\tau 1$ to τk are calculated respectively. These average values $\tau 1'$ to $\tau k'$ are set as the delay times of the system circuit CQT_1 to CQT_k. The delay times SDL_1 to SDL_k are set.

Then, in step S318, the delay times of the delay circuits DLY_1 to DLY_k are adjusted based on the adjust signals SDL_1 to SDL_k corresponding to the delay times $\tau 1'$ to $\tau k'$. Then, the phase characteristic correcting process has been completed.

In this manner, in the phase characteristic correcting process, the loudspeakers are sounded by supplying the noise signal via the system circuits CQT_1 to CQT_k to measure the delay times, and then the phase characteristic is calculated from the sound collecting results of resultant reproduced sounds. Therefore, the delay times of the delay circuits DLY_1 to DLY_k are not simply adjusted (corrected) based on only the propagation delay times of the reproduced sounds, but it is possible to implement the total rationalization while taking account of the reproducing capabilities of respective loudspeakers and the characteristic of the system circuits CQT_1 to CQT_k.

Next, when the phase characteristic correcting process has been completed, the process is shifted to the flatness correcting process in step S40 in FIG.2. The process in step S40 will be carried out in compliance with a flowchart shown in FIG.12.

First, in step S400, the noise signal (uncorrelated noise) DN output from the noise generator 3 can be input by switching the switch elements SW_{11} to SW_{k1}. Also, the amplification factors of the amplifiers 5_{FL} to 5_{WF} are made

equal.

Then, in step S402, the inter-band attenuator ATF_{11} to ATF_{ki}, the channel-to-channel attenuators ATG_1 to ATG_5, and the delay circuits DLY_1 to DLY_k are fixed to their already- adjusted states. However, in step S404, the attenuation factor of the channel-to-channel attenuator ATG_k in the system circuit CQT_k is set to 0 dB.

Then, in step S406, the noise signal (uncorrelated noise) DN is simultaneously supplied to the system circuits CQT_1 to CQT_5 except the system circuit CQT_k. Here, the inter- band attenuators ATF_{11} to ATF_{1i}, ..., ATF_{51} to ATF_{5i} in the low frequency band among the inter-band attenuators ATF_{11} to ATF_{1j}, ..., ATF_{51} to ATF_{5j} in the system circuits CQT_1 to CQT_5 are brought into their OFF (nonconductive) states, and then the above noise signal DN is supplied.

Accordingly, the all frequency band loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR} are simultaneously sounded by the noise signal DN in the middle/high frequency band, then the middle/high frequency band processing portion 15a receives resultant middle/high frequency band sound collecting data D_{MH} (see FIG.4), and then a spectrum average level P_{MH} of the reproduced sounds in the middle/high frequency band by the loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR} is calculated based on the middle/high frequency band sound collecting data D_{MH}.

Then, in step S408, the noise signal (uncorrelated noise)

DN is simultaneously supplied to the system circuits CQT_1 to CQT_5 except the system circuit CQT_k. Here, the inter- band attenuators ATF_{11} to ATF_{1i}, . . . , ATF_{51} to ATF_{5i} in the low frequency band among the inter-band attenuators ATF_{11} to ATF_{1j}, . . . , ATF_{51} to ATF_{5j} in the system circuits CQT_1 to CQT_5 are brought into their ON (conductive) states, and remaining inter-band attenuators are brought into their OFF (nonconductive) states, and then the above noise signal DN is supplied.

Accordingly, the all frequency band loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR} are simultaneously sounded by the noise signal DN in the low frequency band, then the low frequency band processing portion 15b receives resultant low frequency band sound collecting data D_L (see FIG.4), and then a spectrum average level P_L of the reproduced sounds in the low frequency band by the loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR} is calculated based on the low frequency band sound collecting data D_L.

Then, in step S410, the noise signal (pink noise) DN is supplied only to the system circuit CQT_k. Here, the inter-band attenuators ATF_{11} to ATF_{1i}, . . . , ATF_{51} to ATF_{5i} in the low frequency band among the inter-band attenuators ATF_{11} to ATF_{1j}, . . . , ATF_{51} to ATF_{5j} are brought into their ON (conductive) states, and remaining inter-band attenuators are brought into their OFF (nonconductive) states, and then the above noise signal DN is supplied.

Accordingly, only the low frequency band exclusively

reproducing loudspeaker 6_{WF} is sounded by the noise signal DN, then the subwoofer low frequency band processing portion 15c receives resultant subwoofer sound collecting data D_{WFL} (see FIG.4), and then a spectrum average level P_{WFL} of the reproduced sound in the low frequency band reproduced by the loudspeaker 6_{WF} is calculated based on the subwoofer sound collecting data D_{WFL}.

In step S412, the calculating portion 15d calculates the adjust signal SG_k by executing the calculation expressed by following Eq.(10) to adjust the attenuation factor of the channel-to-channel attenuator ATG_k of the system circuit CQT_k.

$$SGk=(TG_L \times P_{MH}-TG_{MH} \times P_L)/(TG_{MH} \times P_{WFL}) \quad \ldots(10)$$

That is, if the audio sound is reproduced by virtue of all loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR}, 6_{WF} by executing the calculation in above Eq.(10), the adjust signal SG_k is calculated to make flat the frequency characteristic of the reproduced sound in the sound field space.

Explaining in detail, the adjust signal SG_k for adjusting the attenuation factor of the channel-to-channel attenuator ATG_k is calculated such that a sum of the level of the reproduced sound in the low frequency band out of the reproduced sound being simultaneously reproduced by the all frequency band loudspeakers 6_{FL}, 6_{FR}, 6_C, 6_{RL}, 6_{RR} and the level of the reproduced sound reproduced by the low frequency band exclusively reproducing subwoofer 6_{WF} is made equal to

the level of the reproduced sound in the middle/high frequency band out of the reproduced sounds that are reproduced simultaneously by the all frequency band loudspeakers 6_{FL}, 6_{FR}, 6_{C}, 6_{RL}, 6_{RR}.

A coefficient TG_{MH} in above Eq.(10) is an average value of the target curve data corresponding to the middle/high frequency band, out of the target curve data which the listener selects among the target curve data [TGxJ] shown in above Eq.(2) or the default target curve data which the listener does not select. Also, a coefficient TG_{L} is an average value of the target curve data corresponding to the low frequency band.

Then, in step S414, the attenuation factor of the channel-to-channel attenuator ATG_{k} is adjusted by using the adjust signal SG_{k}, and then the automatic sound field correcting process has been completed.

In this manner, in the case that the audio sound is reproduced by all frequency band loudspeakers 6_{FL}, 6_{FR}, 6_{C}, 6_{RL}, 6_{RR}, 6_{WF}, the frequency characteristic of the reproduced sound in the sound field space can be made flat over the full audio frequency range if the level correction is executed finally between the channels by the flatness correcting portion 13. Therefore, the problem in the prior art such as the increase of the low frequency band level shown in FIG.6 can be overcome.

Also, in the sound field characteristic measuring process in steps S404 to S410, since the reproduced sounds

generated by sounding respective loudspeakers 6_{FL}, 6_{FR}, 6_{C}, 6_{RL}, 6_{RR}, 6_{WF} on time-division basis are collected, the reproducing capabilities (output power) of respective loudspeakers can be detected. Therefore, the total rationalization with taking the reproducing capabilities of respective loudspeakers into consideration can be achieved.

Then, the audio signals S_{FL}, S_{FR}, S_{C}, S_{RL}, S_{RR}, S_{WF} from the sound source 1 are set into the normal input state by turning OFF the switch element SWN, turning OFF the switch elements SW_{11}, SW_{21}, SW_{31}, SW_{41}, SW_{51}, SW_{k1} connected to this switch element, and turning ON the switch elements SW_{12}, SW_{22}, SW_{32}, SW_{42}, SW_{52}, SW_{k2}, and thus the present audio system is brought into the normal audio playback state.

As described above, according to the present embodiment, since the frequency characteristic and the phase characteristic of the sound field space are corrected while totally taking account of the characteristics of the audio system and the loudspeakers, the extremely high quality sound field space with the presence can be provided.

Also, the problem such that the level of the reproduced sound at a certain frequency in the audio frequency band is increased or decreased, e.g., the problem such that the low frequency band level shown in FIG.6 is increased can be overcome. In other words, since the frequency characteristics of the reproduced sounds being reproduced by respective loudspeakers

is made flat over the entire audio frequency band, such a problem can be overcome that the sound offensive to the ear is produced because the level at the certain frequency is enhanced, and thus the very high quality sound field space with the presence can be implemented.

Also, the correction to implement the very high quality sound field space with the presence is made possible by executing the sound field correcting process in the order of steps S10 to S40 shown in FIG.8.

In addition, since the sound field correction is executed so as to meet to the target curve instructed by the listener, it is possible to improve the convenience, etc.

Further, since the pink noise similar to the frequency characteristic of the audio signal is used in the correction of the frequency characteristic and the correction of the channel-to-channel level and the flattening of level, the correction to meet to the situation that the audio sound is actually reproduced can be achieved with good precision.

In the present embodiment, the automatic sound field correcting system of the so-called 5.1 channel multi- channel audio system that includes the wide frequency range loudspeakers 6_{FL} to 6_{RR} for five channels and the low frequency band exclusively reproducing loudspeaker 6_{WF} has been explained, but the present invention is not limited to this. The automatic sound field correcting system of the present

invention can be applied to the multi-channel audio system that includes the loudspeakers that are larger in number than the present embodiment. Also, the automatic sound field correcting system of the present invention can be applied to the audio system that includes the loudspeakers that are smaller in number than the present embodiment.

That is, the present invention can be applied to the audio system having one or two or more loudspeakers.

The sound field correction in the audio system including the low frequency band exclusively reproducing loudspeaker (subwoofer) 6_{WF} has been explained, but the present invention is not limited to this. The high quality sound field space with the presence can be provided by the audio system that includes the high frequency band exclusively reproducing loudspeaker and the all frequency band loudspeakers, or the audio system that includes the low frequency band exclusively reproducing loudspeaker, the high frequency band exclusively reproducing loudspeaker, and the all frequency band loudspeakers.

In the present embodiment, in step S412 shown in FIG.12, as apparent from above Eq.(10), the rationalization of the attenuation factor of the channel-to-channel attenuator ATG_K is performed on the basis of the levels of the reproduced sounds of all frequency band loudspeakers 6_{FL} to 6_{RR}. That is, the levels of the reproduced sounds of all frequency band

loudspeakers 6_{FL} to 6_{RR} are used as the basis by setting a product of the target data TG_{MH} in the middle/high frequency band and the variable P_{WFL}, that corresponds to the spectrum average level of the reproduced sound of the low frequency band exclusively reproducing loudspeaker 6_{WF}, in the denominator of above Eq. (10). However, the present invention is not limited to this. The rationalization of the attenuation factors of the channel-to-channel attenuators ATG_1 to ATG_5 is performed on the basis of the level of the reproduced sound of the low frequency band exclusively reproducing loudspeaker 6_{WF}.

That is, in the present embodiment, the flatness correcting portion 14 corrects the attenuation factor of the channel-to-channel attenuator ATG_K. Conversely, the level of the reproduced sound of the low frequency band exclusively reproducing loudspeaker 6_{WF} may be measured, then the attenuation factor of the channel-to-channel attenuator ATG_K may be set on the basis of measured result, and then the attenuation factors of the channel-to-channel attenuators ATG_1 to ATG_5 may be corrected on the basis of the attenuation factor of the channel-to-channel attenuator ATG_K.

Further, as described above, the system circuits CQT1 to CQTk shown in FIG.2 is constructed by connecting the band-pass filters, the inter-band attenuators, the adder, the channel-to-channel attenuator, and the delay circuit in sequence. However, such configuration is shown as the typical

example and thus the present invention is not limited to such configuration.

For example, the delay circuit that is connected following to the channel-to-channel attenuator may be arranged on the input side of the band-pass filters or the input side of the inter-band attenuators. Also, the positions of the channel-to-channel attenuator and the delay circuit may be exchanged. In addition, both the channel-to-channel attenuator and the delay circuit may be arranged on the input side of the band-pass filters.

The reasons for enabling the configuration of the present invention to change appropriately the positions of the constituent elements are that, unlike the conventional audio system in which the correction of the frequency characteristic and the correction of the phase characteristic are performed respectively by separating respective constituent elements, the noise signal from the noise generator can be input from the input stage of the sound field correcting system and also the frequency characteristic and the phase characteristic of the overall sound field correcting system can be corrected totally. As a result, the automatic sound field correcting system of the present invention makes it possible to correct properly the frequency characteristic and the phase characteristic of the overall audio system and to enhance margin in design.

As described above, according to the sound field correcting method of the present invention, when the audio signals are reproduced by the sound generating means (loudspeakers) having different reproducing frequency bands, the levels of the reproduced sounds reproduced by respective sound generating means can be made flat over the entire reproducing frequency band. As a result, the high quality sound field space with the presence can be provided.